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SIPp: Enhancement in SIP user Agents Emulation

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Abstract

VOIP is widely used Protocol for IP Telephony related products. SIP is text based VOIP protocol used for establishing sessions. SIP proposed by IETF and being used in many IP Telephony Products. PBX is Private Branch Exchange used for switching calls between local network and public network. IP-PBX is IP based PBX that comes up multiple feature with call management. The IP-PBX need to be tested for its features like conferencing, call forwarding, etc. SIPp is load testing tool used for testing features of IP-PBX. Here we have used Asterisk to handle load. Proposed approach is generation of hybrid load that will generate load consisting multiple scenarios and which was not possible by SIPp as SIPp generates load of only single scenario.

Keywords: Private Branch Exchange(PBX), Session Initiation Protocol(SIP), Voice over Internet Protocol(VoIP).

Introduction

The requirements of multimedia features in Communication Technology are rapidly increasing. The respective systems should be capable of having such features and able to handle load with these features. As generic PSTN network cannot prove itself to be capable enough in satisfying recent needs in Communication, hence VOIP came into the picture to suffice multimedia needs. VOIP is fast growing and mostly used technology in current industrial scenario. RFC 6405 represents VOIP which is a transfer of digitized voice in form of packets. VOIP mainly consist of two main Protocols, these are SIP and H.323.[2]Current paper involves SIP based telephony Systems. SIP follows RFC 3261 that is a Signaling Protocol. SIP can establish, Control and Terminate connection. SIP handles Media sessions using RTP and to describe media sessions it uses Session Description Protocol. To test SIP Based application it requires a tool which can test features and load handling capability of application. There are many open source tool for testing SIP like, SIPp, SIPsak, Seagull. Out of which SIPp is reliable and efficient tool for testing SIP applications.[3] Asterisk is open source tool performs as IP PBX. Asterisk is capable of providing multimedia telephony features like Voicemail, Automatic attendant, Call Parking, Paging, Conference Bridging, Intercomm Calling. Moreover it also supports different kinds of trunks by which VoIP Provides Unified Communication.[1] Proposed architecture focuses on effective generation of load to stress the system. Effective load generation refers to the hybrid load generation that will stress system to get capability.

Session Initiation Protocol (SIP)

Session Initiation Protocol is a Signaling Protocol used for Establishing, Controlling and Terminating the session.SIP is designed in 1996 by Henning Schulzrinne and Mark Handley. Afterward it is standardized by Internet Engineering Task Force as RFC 3261. SIP call is controlled using SDP that helps maintaining details of call(i.e., audio, video, codec type, size of packet, etc.). SIP works without underlying dependency of transport protocol and session type that is being established. SIP is text based Application Layer Protocol of OSI(Open System Interconnection) Model. SIP involves many elements of HTTP(Hypertext Transfer protocol) and SMTP(Simple Mail Transfer Protocol). Establishing Multimedia session includes RTP, RTSP, MEGACO, and SDP. While SIP is a component that is used with IETF Protocols.[3]

SIP works by enabling its endpoints that is User Agents. These End Points can be User Agent server and User Agent Client.

1. User Agent Client(UAC) is endpoint which requests the server.
2. User Agent Server(UAS) it handles requests and responds accordingly.

IP-Telephone is a client that will generate request and send it to server. Server will act according to its own responsibility. According to servers requirement in Network, there are several kinds of servers in SIP as follows:

1. Proxy Server acts at both ends that contribute in making requests and giving responses.
2. Redirect Server has job of redirecting requests in case of different route requirement.
3. Register Server registers users.
4. Location Server keeps address of registered users.

SIP Requests and Responses are two different kinds of SIP Messages. Following Message Types comes under SIP Requests.

SIP Requests

1. REGISTERregisters the current location of user.
2. INVITE forms session between users.
3. ACK that confirms reliable message exchange.
4. CANCEL terminates pending requests.

5. BYE terminates session between users.
6. OPTIONS requests information about user Capabilities.

Following SIP Messages comes under SIP Responses.

SIP Responses

1. XX(Provisional) implies request is received and processed.
2. XX(Success) implies action completely received and accepted by Server.
3. XX(Redirection) further action to be taken to complete requests.
4. XX(Client Error) implies bad request has been made and cannot processed.
5. XX(Server Error) implies server failed to fulfill the request made.
6. XX(Global Failure) implies request cannot be fulfilled by any server.

Asterisk IP PBX

Asterisk is a free and Open Source framework which is used for building Telephony Applications. Asterisk is developed by Mark Spencer in 1999, as a software implementation of Internet Protocol-Private Branch Exchange (IP-PBX). Voicemail, Conference, Interactive Voice Response (IVR) and Automatic Call Distribution are supported by Asterisk. Asterisk supports SIP (Session Initiation Protocol), H.323, Inter-Asterisk Exchange and MGCP (Media Gateway Control Protocol). [1]

Asterisk can be used on variety of Linux distributions. Asterisk used by smaller to midsize businesses. Asterisk benefits both developers and researchers to develop IP Telephony applications and test them.[1]

SIPp

SIPp is a load and feature testing tool for IP Telephony Applications. It simulates the load and deploys it on IP Telephony application such as Asterisk so that tester measures the capacity and performance of the application. Following are main features that makesSIPp unique:

1. SIPp allows generation of multiple SIP calls from single remote system.
2. SIPp has integrated Scenarios. These are
   a. UAC
   b. UAC with media
   c. UAS
   d. Regexp
   e. Branch
   f. UAC out of call message
   g. 3PCC (3rd Party Call Control)

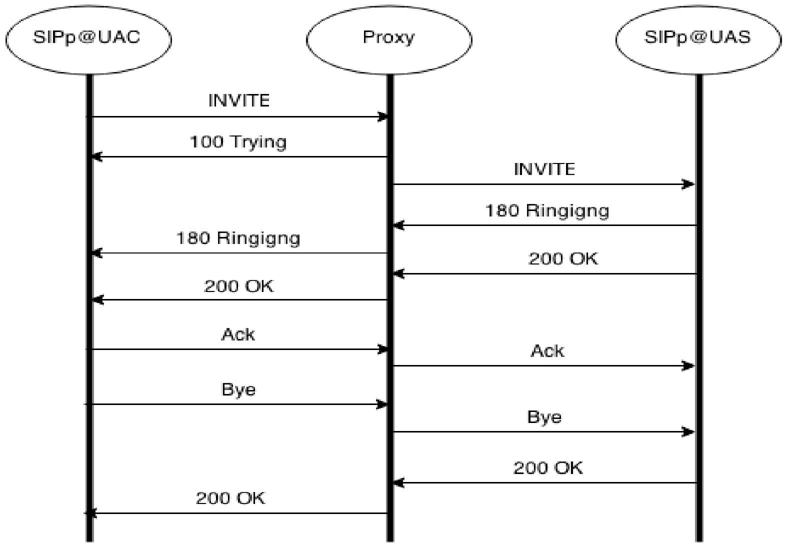

Fig. Performance Test

Related Work: (Asterisk, SIP sak, Seagull)

There are three major organizations that are hot in providing IP Telephony solutions. Organizations those have covered big market in IP-Telephony are Avaya, Cisco, Microsoft and more are rising with time. The work around in Load Testing of IP-PBX covers tools that are currently used for testing load and feature of IP Based Telephony products. We have used open source IP-PBX i.e. Asterisk. Asterisk is used to check the affectivity of multiple load testing tools. These tools are SIPp, Seagull, Sipsak.

Sipsak

Sipsak is Command based tool used by developers and administrators of SIP applications. It can be used by testers for checking the capabilities of SIP Applications.[5]

Features

1. Sending options request.
2. Sending text files containing SIP Requests.
3. Trace route.
4. User Location test.
5. Flooding test.
6. Random character trashed test.
7. Interpret and react on response.
8. String replacement in files.
9. Can simulate calls in usrloc mode.
10. Uses symmetric signaling and thus should work behind NAT.

Seagull

Seagull is open source tool for testing IP Telephony applications released under the terms of GNU GPL V2 license. Seagull is an effective testing tool mainly designed for IMS related applications. It is traffic generator for functional, load, endurance, stress and performance check. Seagull comes up with several brand protocol families that it has used in its source code. I.e. Binary/TLV (Diameter, Radius, IETF and 3GPP protocols), External Library (TCAP,PCP), Text (XCAP, HTTP, H248 ASCII)[6].

Seagull is traffic generator tool works on Multi-protocols developed by hp. Seagull is developed using C++ and uses XML to create scenario. Scenario describes the messages that are to be sent and received.

Features:

1. Traffic Generation for multiple Protocols.
2. Command Line Tool having text interface.
3. Scenario Description using XML file.
4. Support of IP, (UDP/TCP), SCTP, SSL/TLS and SS7/TCAP transports.
5. Multi-Threaded in Performance and Reliability.
6. Remote Control through standard HTTP interface.
7. Multiple scenario synchronization in meddle of scenario.
8. Scenario Display with message counters.

Primary aim of Seagull is towards IMS Protocols, while SIPp is limited only to SIP Protocol. Seagull is performance testing tool for IMS application and SIPp is performance testing tool for SIP based IP Telephony applications.[6]

We have chosen SIPp tool for improving the test over PBX. As SIPp take XML input containing scenario description which gives flexibility in designing scenarios. So it makes simple in providing input to the system. SIPp fully implements the RFC 6076 metrics which helps to generate all possible call scenarios. Moreover SIPp helps to successfully stress the System to get desired results.

Hybrid Load Generation

Since SIPp is single threaded tool which generates single scenario at a moment. It is difficult for SIPp to generate load consisting multiple scenarios. This implies SIPp has limited capability in testing the PBX effectively.The proposed approach is generation of hybrid load. In which the SIPp load can be combined and then applied to PBX. The new Setup is having addition of extra component that is load handler. Load handler stays responsible for combining multiple load scenarios and assigns that load to PBX to check capability of PBX to perform. Load Handler or Load Scheduler requires a scheduling algorithm for scheduling the type of loads and give them as input for PBX. Here we have used Round Robin scheduling algorithm in assigning load to the PBX. PBX will get sequential assignment of load and thus the check of performance.

Architecture

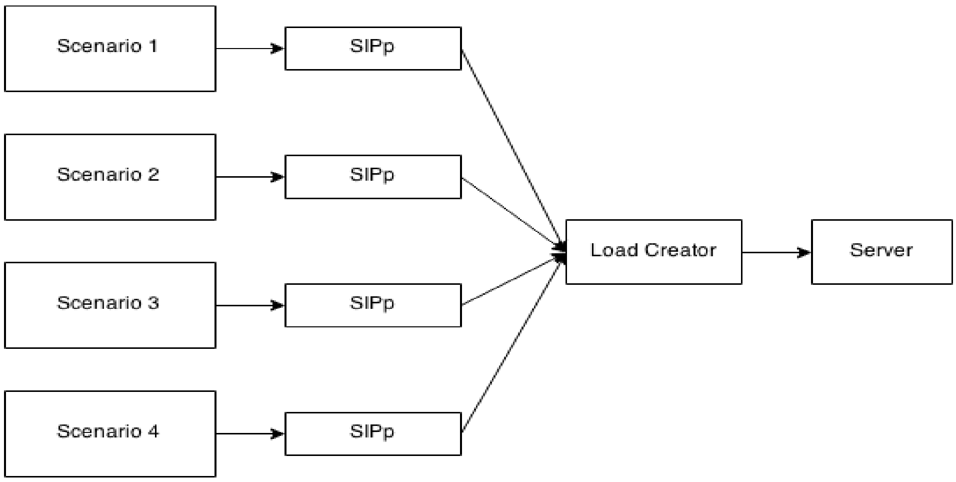

Fig : Block Diagram

Proposed System

Proposed system is designed as per tester's requirements such as ease of selecting scenario, selection of PBX, Generation of output on console or in Document. To provide ease of test suite generation is the main purpose of desired task. The system is developed using Java. The APIrelated to SIPphas been used. Since Java is Object Oriented language and it is multithreaded that had helpeddeveloping the system more effectively.

The output of the system is in the form of Report. That has report of generated scenarios and record of PBX performance. Generated output has two components i.e. UAC Screen and UAS Screen. Input section has list of scenarios that are injected for generation of load with respective count. Output section has all the details regarding call flow and time taken to get response from PBX.

Results

The Results of the system contains 2 elements in single section while elements are SIPp command line output and logs generated. One section shows Client command line console and second section shows the response from server. Second section represents server side that contains amount of requests coming reply to those requests. At server section theresponse givento each client is being shown. Server screen will represent the amount of time it has taken to execute the scenarios.Previously the PBX testinused to perform by generating load of single scenario at moment. Now we are able to give load consisting more than single scenario to single PBX that makes PBX checking more effective. Implementation of Round Robin Algorithm required inclusion of extra element that is Load Creator. Proposed architecture affected due to load creator indicating less efforts for tester to test utter testing of PBX. The use of proposed approach has reduced Load Testing time. Sometimes it depends on Scenarios too.

Conclusion

Load testing of IP-PBX have numerous availability of tools. Every other tool has its different methods to do load testing. SIPp is reliable and provides more functionality in load and feature testing of SIP based IP Telephony Application. However SIPp also has its own limitation of being single threaded. Due to which providing load of more than single scenario is not possible. That is being eliminated by scheduling the load using Round Robin Algorithm. This Improvement has caused time saving approach in case of multiple scenarios are to be given to particular PBX. Hence this approach saves time and efforts of tester in load testing of SIP Based IP-PBX.

Future Scope

In this paper we have scheduled the scenarios using Round Robin Algorithm. Implementation of different scheduling algorithm may produce more effective results. The processing time of load may reduce by using different scheduling algorithm.

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